UNIT–03

UNIT-03/LECTURE-01

DIGITAL CONVERSION OF ANALOG SIGNALS

The Sampling Theorem:- **(DEC 2013)(7)**

The sampling theorem provides the basis for transmitting analog signals by use of digital techniques.

The sampling theorem may be stated in two equivalent way as under:

(i) A band limited signal having no frequency component higher than fm Hz is completely described by its sample values at uniform intervals less than or equal to 1/2fm second apart.
This is frequency domain statement.

(ii) A band limited signal having no frequency component higher than fm Hz may be completely recovered from the knowledge of its samples taken at the rate of at least 2fm samples per second.

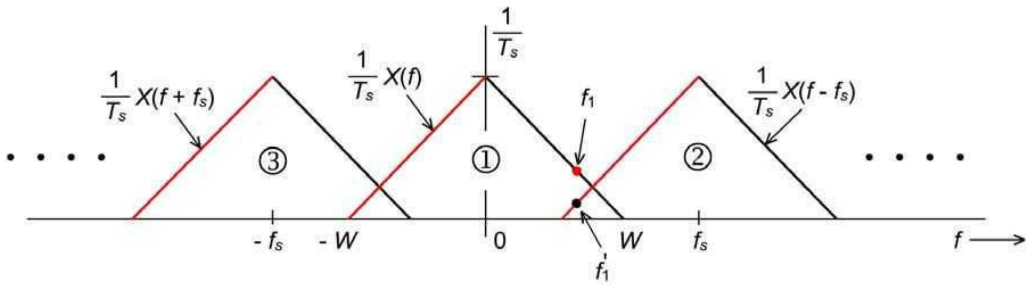
This is time domain statement

Sampling and interpolation take us back and forth between discrete and continuous time and vice versa. However - our reconstructed (interpolated) continuous time signal is by no means guaranteed to be even close to the original continuous time signal. A major breakthrough for doing this sampling and interpolation business 'right' was achieved by Claude Shannon in 1948 with his famous Sampling Theorem.

The sampling theorem states conditions under which a continuous time signal can be reconstructed exactly from its samples and also defines the interpolation algorithm which should be used to achieve this exact reconstruction.

Aliasing:-

When fs< 2wm (2fm) , we have seen that there is spectral overlap and the resulting distortion is called the aliasing distortion



Sampling of Non-bandlimited Signal: Anti-aliasing Filter:-

Anti aliasing filter is a filter which is used before a signal sampler, to restrict the bandwidth of a signal to approximately satisfy the sampling theorem. The potential defectors are all the frequency components beyond fs/2 Hz. We should have to eliminate these components from x(t) before sampling x(t). As a result of this we lose only the components beyond the folding frequency fs/2 Hz. These frequency components cannot reappear to corrupt the components with frequencies below the folding frequency. This suppression of higher frequencies can be accomplished by an ideal filter of bandwidthfs/2 Hz. This filter is called the anti-aliasing filter. The anti aliasing operation must be performed before the signal is sampled. The anti aliasing filter, being an ideal filter is unrealizable. In practice, we use a steep cutoff filter, which leaves a sharply attenuated residual spectrum beyond the folding frequency fs/2.

Signal Reconstruction:-

The process of reconstructing a continuous time signal x(t) from its samples is known as interpolation. In the sampling theorem we saw that a signal x(t) band limited to D Hz can be reconstructed from its samples. This reconstruction is accomplished by passing the sampled signal through an ideal low pass filter of bandwidth D Hz. ,the sampled signal contains a component and to recover x(t) or X(f), the sampled signal must be passed through an ideal lowpass filter having bandwidth D Hz and gain T.

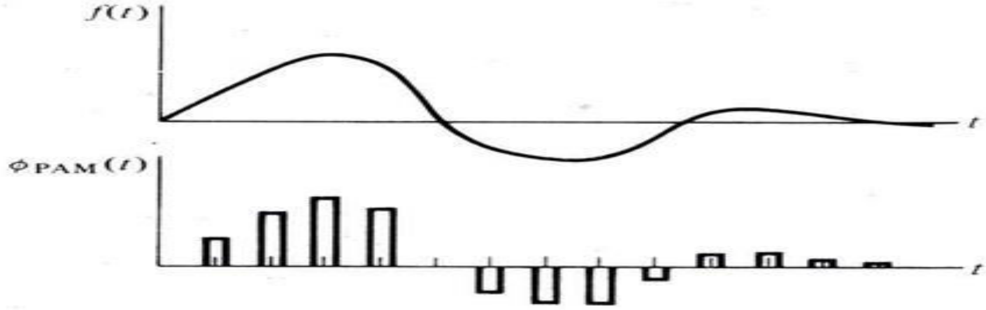
	RGPV QUESTIONS	Year	Marks
Q.1	State and prove sampling theorem.	DEC 2013	7
Q.2	State and prove sampling theorem?	JUNE2013	7
Q.3	State and prof. sampling theorem. What is aliasing explain.	DEC 2012	10

UNIT-03/LECTURE-02

PULSE AMPLITUDE MODULATION

Pulse amplitude modulation (PAM):- **(DEC 2013)(7)**

PAM sampling the amplitude of a train of constant width pulses is varied in proportion to the sample values of the modulating signal as shown below



PULSE AMPLITUDE MODULATION

Pulse Modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The Pulse Amplitude Modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable are used to module division multiplexing is used.

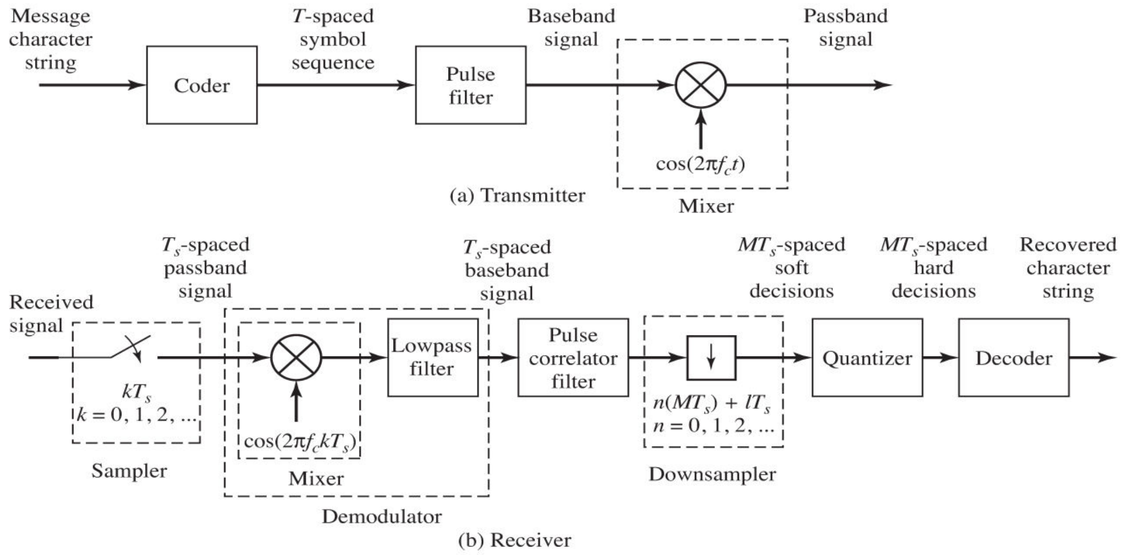


Fig: PAM Modulator and Demodulator

	RGPV QUESTIONS	Year	Marks
Q.2	Draw and explain the circuit of PAM modulator and demodulator	DEC 2013	7
Q.1	Draw and explain the PAM modulator and demodulator circuit?	DEC 2011	10

UNIT-03/LECTURE-03

TYPE OF SAMPLING

<u>Types of sampling</u>:- **(DEC 2013)(7)**

There are three sampling types available, these are

1. Ideal sampling
2. Natural sampling
3. Flat top sampling

Ideal sampling

In ideal sampling the analog signal is multiplied by a delta comb functions as shown in Fig.
Ideal sampling is used to explain the main concept of sampling theoretically
In practical life Ideal sampling cannot be achieved, because there is no practical circuit which generates exact delta comb function

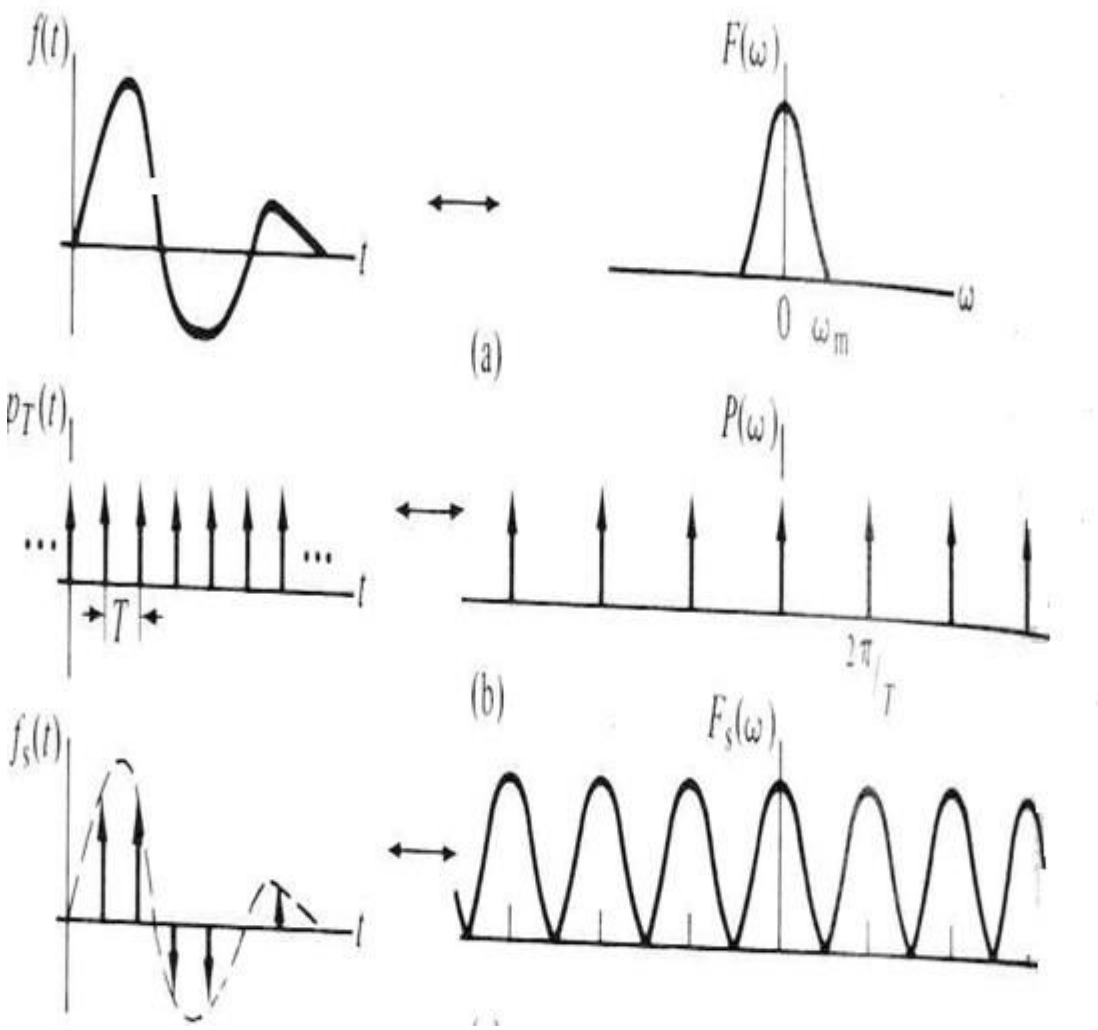


Fig: Ideal sampling

Natural sampling

In natural sampling the information signal f(t) is multiplied by a periodic pulse train with a finite pulse width τ as shown below.

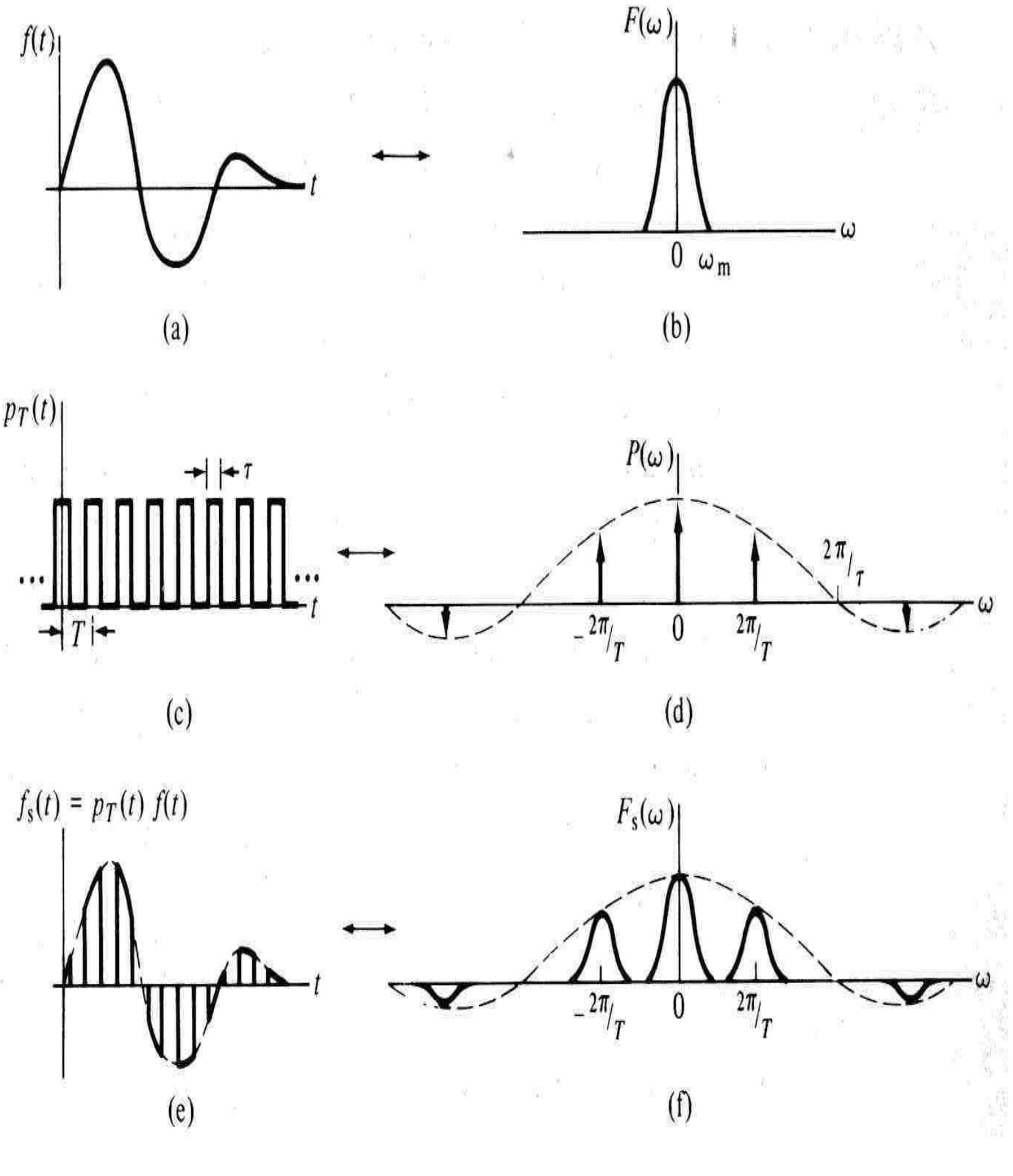


Fig: Natural sampling

As shown in the figure, the natural sampling produces rectangular pulses whose amplitude and top curves depends on the amplitude and shape of the message signal f(t)

Flat top sampling

In flat top sampling the amplitude of a train of constant width pulses is varied in proportion to the sample values of the modulating signal as shown below, In PAM, the pulse tops are flat.PAM Is a example of flat top sampling,

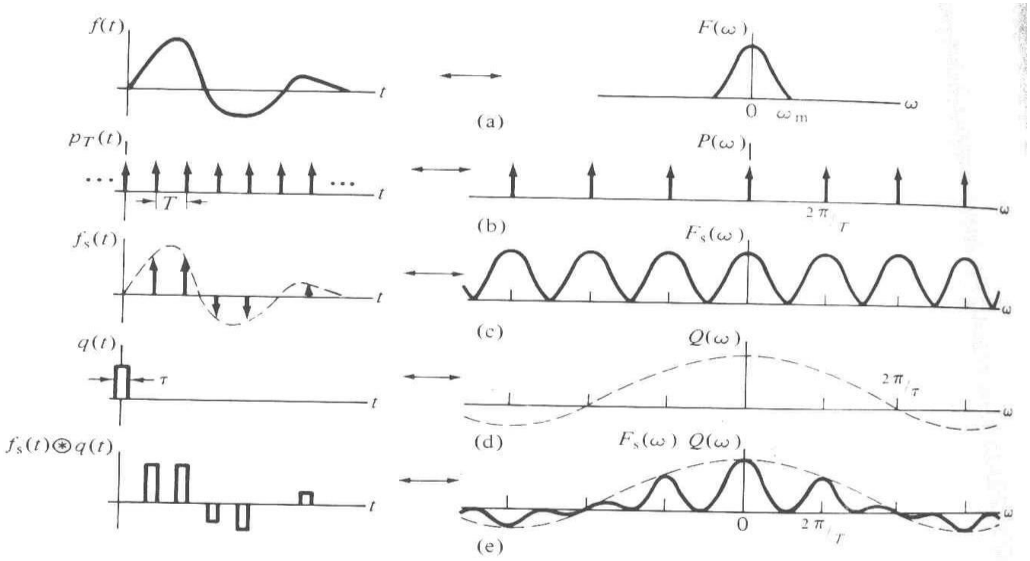


Fig: flat top sampling

	RGPV QUESTIONS	Year	Marks
Q.1	Explain flat top and natural sampling. Why flat top sampling preferred over natural sampling?	DEC 2013	7
Q.2	Explain Flat topped sampling and aperture effect.	DEC 2012	7
Q.3	Explain natural and flat top sampling. Compaire the two. Also describe aperture effect.	JUNE 2012	7
Q.4	Distinguish between instantaneous sampling, natural sampling and flat top sampling. With the functional block diagram explain the working of a circuit that provides flat top sampling.	DEC 2011	10
Q.5	State & explain Sampling theorem with flat-top sampling technique.	JUNE 2010	10
Q.6	Explain natural sampling and flat top sampling, compare the two.	JUNE 2009	10

UNIT-03/LECTURE-04

PULSE TIME MODULATING (PWM, PPM)

<u>Pulse Time Modulation</u>:- **(DEC 2013)(7)**

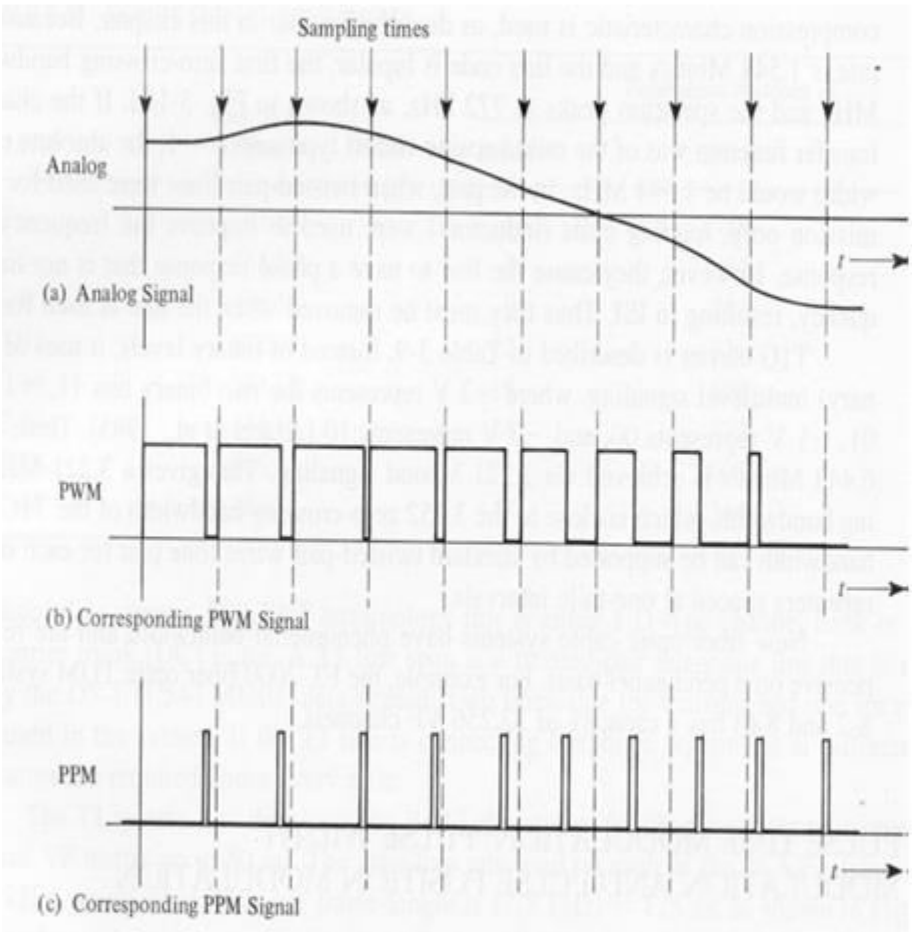
Pulse Width Modulation& Pulse Position Modulation Pulse Time Modulation (PTM) is a class of signaling technique that encodes the sample values of an analog signal onto the time axis of a digital signal.

The two main types of pulse time modulation are:

1. Pulse Width Modulation (PWM)
2. Pulse Position Modulation (PPM)

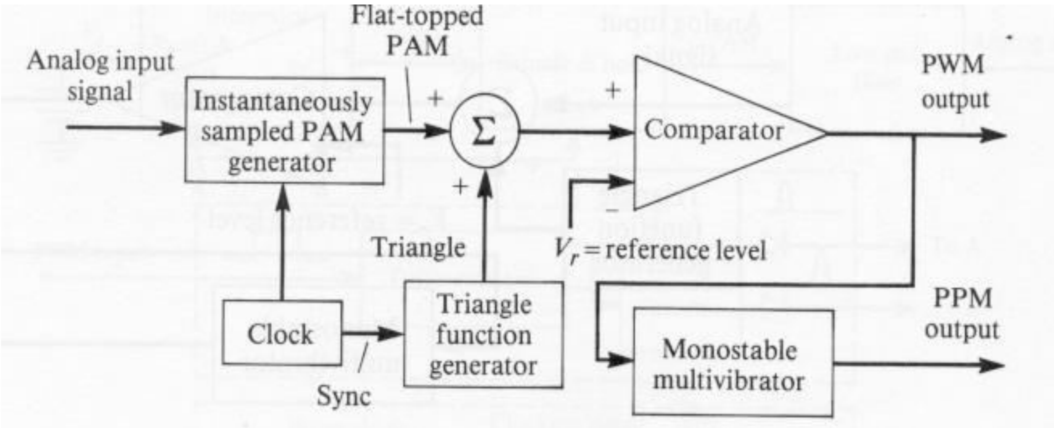
In PWM the sample values of the analog waveform are used to determine the width of the pulse signal. Either instantaneous or natural sampling can be used.

In PPM the analog sample values determine the position of a narrow pulse Relative to the clocking time. It is possible to obtain PPM from PWM by using a mono-stable multivibrator circuit.



PULSE TIME MODULATING SIGNALIND

Figure below shows PWM generation using instantaneous sampling



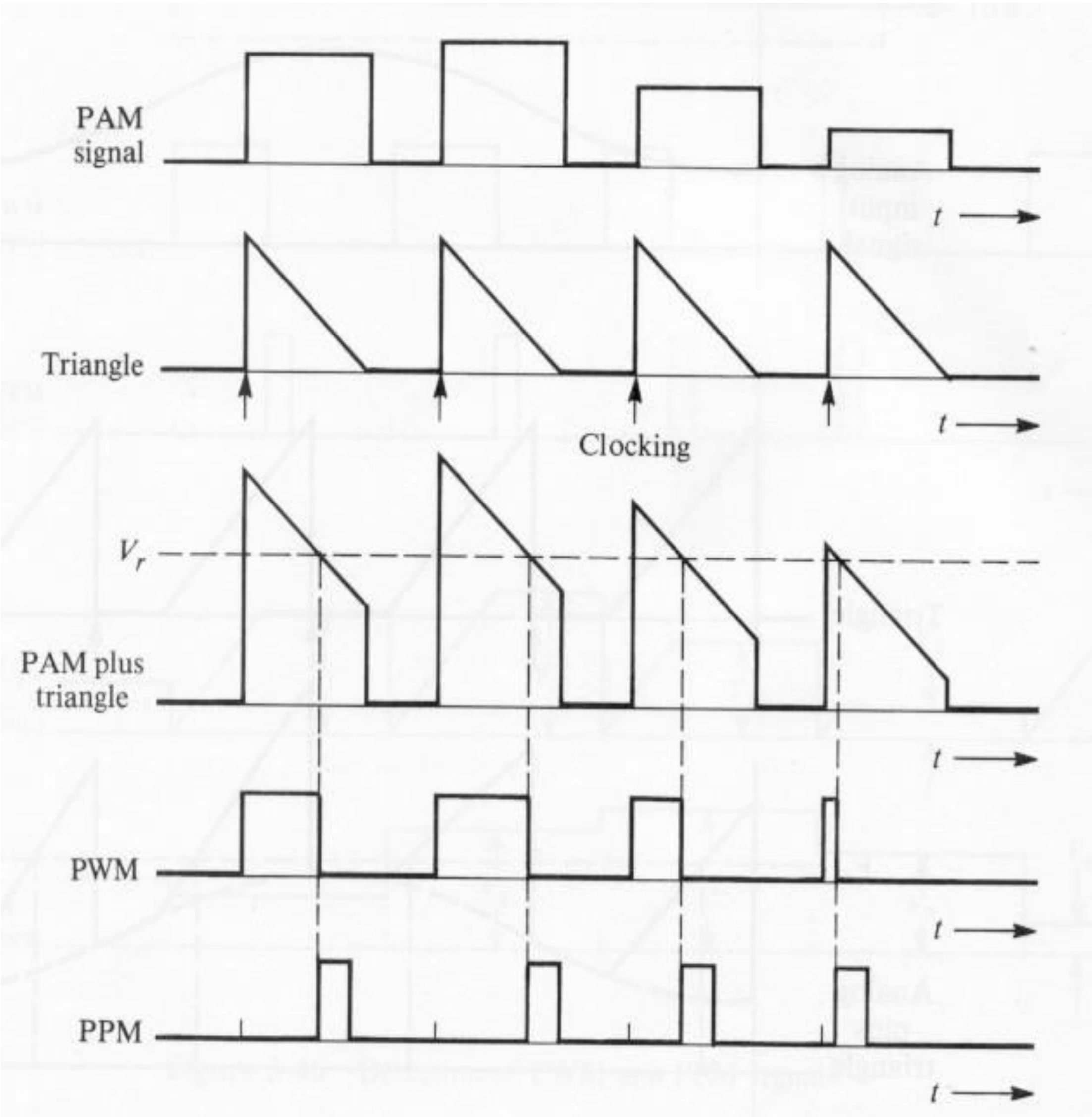


Fig: Techniques for generating instantaneously sampled PTM signals.

Figure shows PWM signal generation using natural sampling.

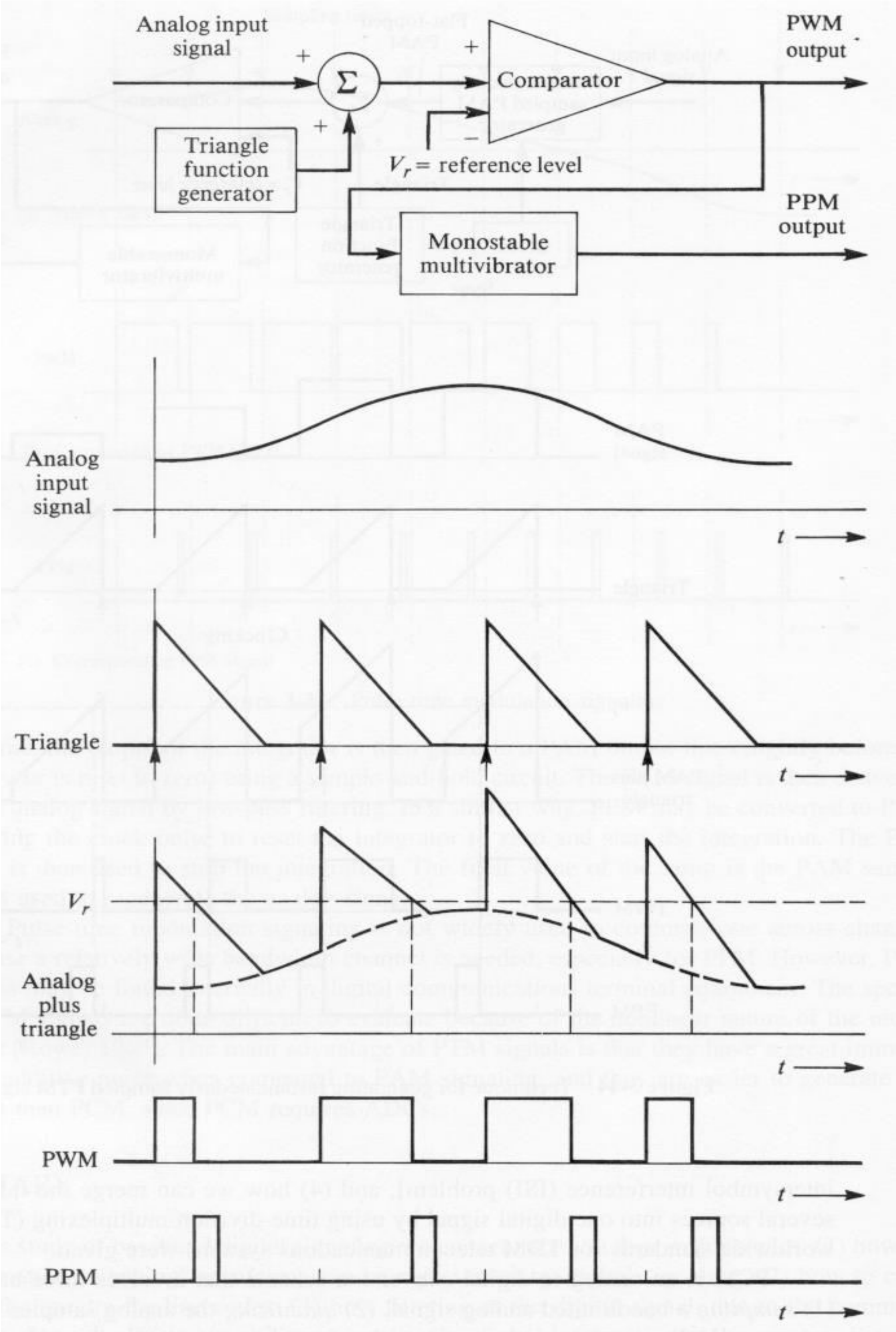


Fig: Techniques for generating naturaly sampled PTM signals.

The PWM or PPM signals may be converted back to the corresponding analog signal by a receiving system as shown in Fig.

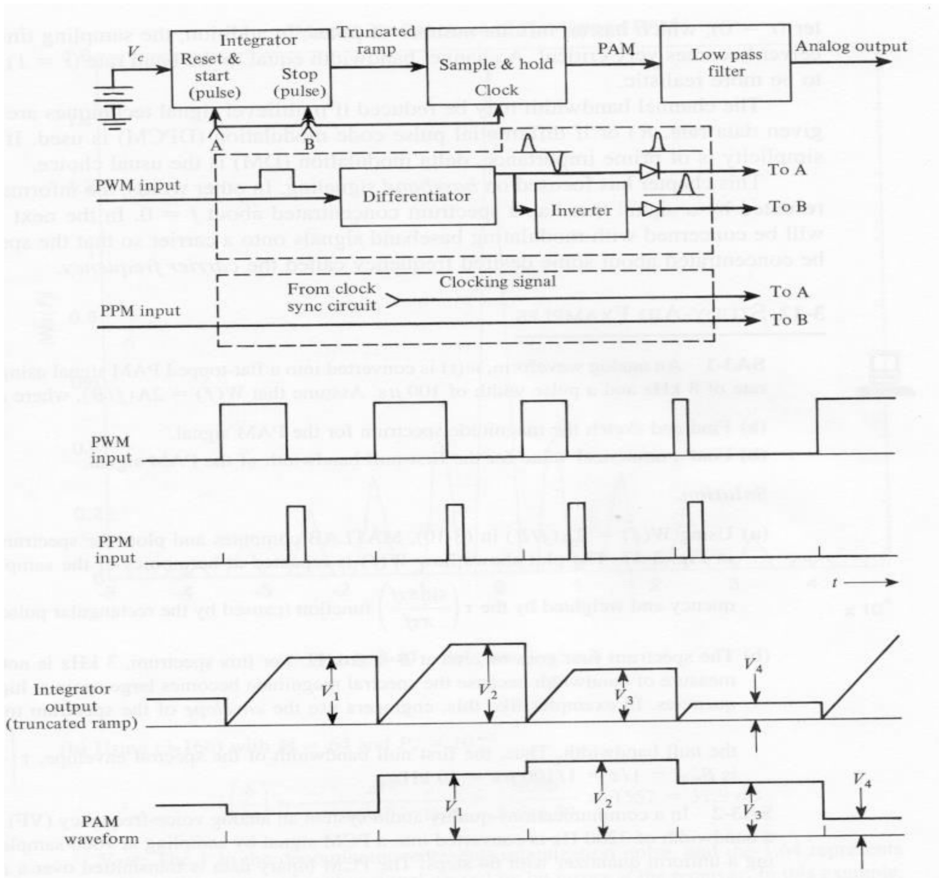


Fig: detection of PWM and PPM signals.

For PWM detection the PWM signal is used to start and stop the integration of the integrator. After reset integrator starts to integrate during the duration of the pulse and will continue to do so till the pulse goes low.

If integrator has a DC voltage connected as input, the output will be a truncated ramp. After the PWM signal goes low, the amplitude of the truncated ramp will be equal to the corresponding PAM sample value. Then it goes to zero with reset of the integrator.

	RGPV QUESTIONS	Year	Marks
Q.1	Explain how PPM and PWM signals are generated 1) from PAM signal 2) directly how are these detected ?	DEC 2013	7
Q.2	Explain. How PPM and PWM signal are generated from PAM signal and how these are detected.	DEC 2012	10

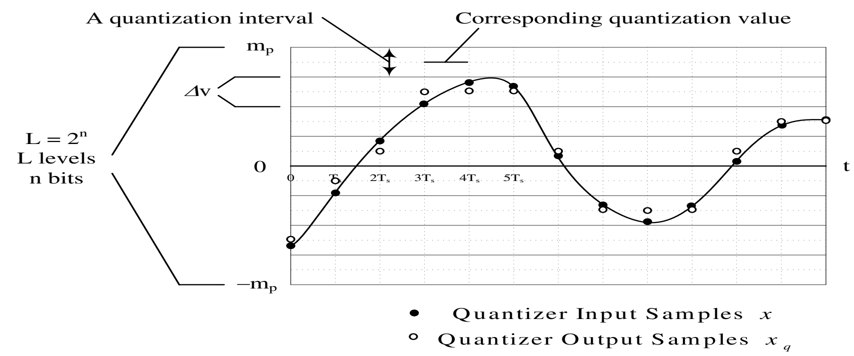
UNIT-03/LECTURE-05
QUANTIZATION

QUANTIZATION:- **(DEC 2013)(7)**

The process of quantizing a signal is the first part of converting an sequence of analog samples to a PCM code. In quantization, an analog sample with an amplitude that may take value in a specific range is converted to a digital sample with an amplitude that takes one of a specific pre–defined set of quantization values. This is performed by dividing the range of possible values of the analog samples into L different levels, and assigning the center value of each level to any sample that falls in that quantization interval.

The problem with this process is that it approximates the value of an analog sample with the nearest of the quantization values. So, for almost all samples, the quantized samples will differ from the original samples by a small amount. This amount is called the quantization error. To get some idea on the effect of this quantization error, quantizing audio signals results in a hissing noise similar to what you would hear when play a random signal.

Assume that a signal with power $P_s$is to be quantized using a quantizer with $L = 2^n$ levels ranging in voltage from $-m_p$ tom_p as shown in the figure below.

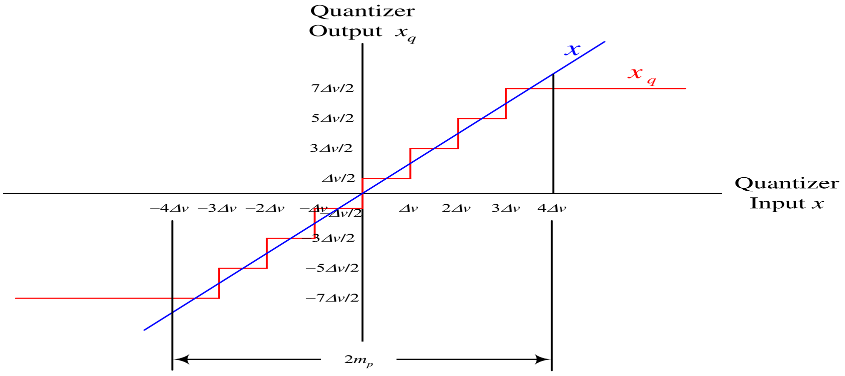


We can define the variable Δv to be the height of the each of the L levels of the quantizer as shown above. This gives a value of Δv equal to

$$\Delta v = \frac{2m_p}{L}.$$

Therefore, for a set of quantizers with the same m_p, the larger the number of levels of a quantizer, the smaller the size of each quantization interval, and for a set of quantizers with the same number of quantization intervals, the larger m_p is the larger the quantization interval length to accommodate all the quantization range.

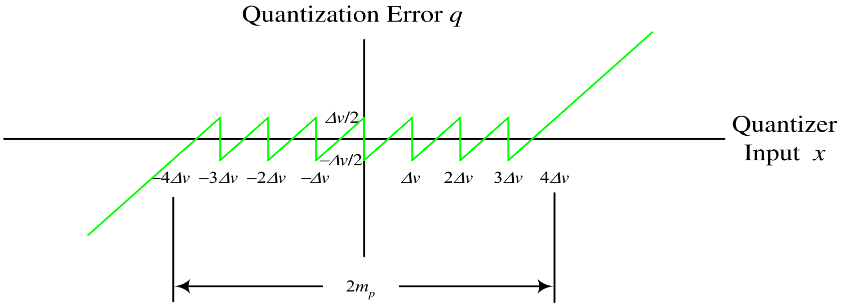
Now if we look at the input output characteristics of the quantizer, it will be similar to the red line in the following figure. Note that as long as the input is within the quantization range of the quantizer, the output of the quantizer represented by the red line follows the input of the quantizer. When the input of the quantizer exceeds the range of $-m_p$ to m_p, the output of the quantizer starts to deviate from the input and the quantization error (difference between an input and the corresponding output sample) increases significantly.



Now let us define the quantization error represented by the difference between the input sample and the corresponding output sample to be q, or

$$q = x - x_q .$$

Plotting this quantization error versus the input signal of a quantizer is seen next. Notice that the plot of the quantization error is obtained by taking the difference between the blow and red lines in the above figure.



It is seen from this figure that the quantization error of any sample is restricted between $-\Delta v/2$ and $\Delta v/2$ except when the input signal exceeds the range of quantization of $-m_p$ to m_p.

	RGPV QUESTIONS	Year	Marks
Q.1	Why quantization required ? Explain quantization error in detail	DEC 2013	7
Q.2	Explain quantaization what is quantaization error. how it depend on step size?	DEC 2012	10
Q.3	Explain quantization. What is quantization error ? How does it depend upon the step size ? explain	JUNE 2012	7

UNIT-03/LECTURE-06

PULSE CODE MODULATION

Pulse Code Modulation (PCM):- **(DEC 2013)(7)**

The modulation methods PAM, PWM, and PPM discussed in the previous lecture still represent analog communication signals since the height, width, and position of the PAM, PWM, and PPM, respectively, can take any value in a range of values. Digital communication systems require the transmission of a digital for of the samples of the information signal. Therefore, a device that converts the analog samples of the message signal to digital form would be required. Analog to Digital Converters (ADC) are such devices. ADCs sample the input signal and then apply a process called quantization. The quantized forms of the samples are then converted to binary digits and are outputted in the form of 1's and 0's. The sequence of 1's and 0's outputted by the ADC is called a PCM signal (Pulses have been coded to 1's and 0's).

Two basic operations in the conversion of analog signal into the digital is time discretization and amplitude discretization. In the context of PCM, the former is accomplished with the sampling operation and the latter by means of quantization. In addition, PCM involves another step, namely, conversion of quantized amplitudes into a sequence of simpler pulse patterns (usually binary), generally called as code words. (The word code in pulse code modulation refers to the fact that every quantized sample is converted to an R -bit code word.)

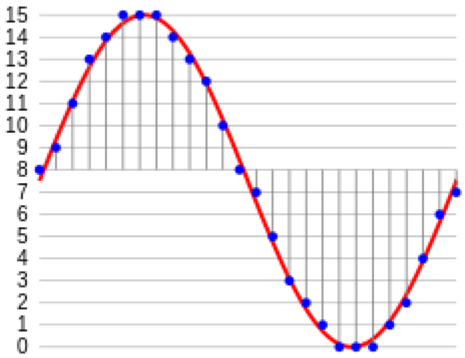


Fig. Sampling and quantization of a signal (red) for 4-bit PCM

Here m(t) is a message signal that is to be transmitted digitally. m(t) is first sampled and then quantized. T is the sampling period and n is the appropriate integer. Fs=1/fm is called the sampling rate or sampling frequency values that is closest to it from among a pre-selected set of discrete amplitudes. The encoder represents each one of these quantized samples by an R -bit code word. This bit stream travels on the channel and reaches the receiving end. With f as the sampling rate and R -bits per code word, the bit rate of the PCM system is R/ts.
The decoder converts the R -bit code words into the corresponding (discrete) amplitudes. Finally, the reconstruction filter, acting mt.

Demodulation

To recover the original signal from the sampled data, a "demodulator" can apply the procedure of modulation in reverse. After each sampling period, the demodulator reads the next value and shifts the output signal to the new value. As a result of these transitions, the signal has a significant amount of high-frequency energy caused by aliasing. To remove these undesirable frequencies and leave the original signal, the demodulator passes the signal through analog filters that suppress energy outside the expected frequency range (greater than the Nyquist frequency $f_s/2$).The sampling theorem shows PCM devices can operate without introducing distortions within their designed frequency bands if they provide a sampling frequency twice that of the input signal. For example, in telephony, the usable voice frequency band ranges from approximately 300 Hz to 3400 Hz. Therefore, per the Nyquist–Shannon sampling theorem, the sampling frequency (8 kHz) must be at least twice the voice frequency (4 kHz) for effective reconstruction of the voice signal.

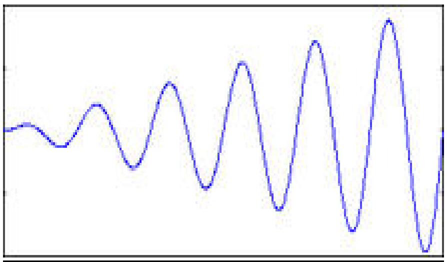
The electronics involved in producing an accurate analog signal from the discrete data are similar to those used for generating the digital signal. These devices are Digital-to-analog converters (DACs). They produce a voltage or current (depending on type) that represents the value presented on their digital inputs. This output would then generally be filtered and amplified for use.

	RGPV QUESTIONS	Year	Marks
Q.1	What are the various process involved in PCM ? With the help of block diagram explain the working of PCM transmitter and receiver ?	DEC 2013	7
Q.2	Explain the block diagram of PCM System. Differentiate Between DM & PCM.	DEC 2010	10

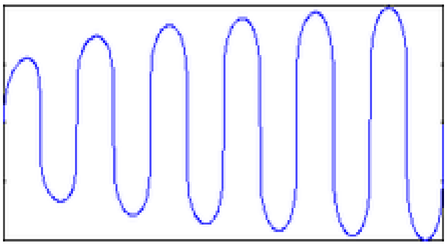
UNIT-03/LECTURE-07
COMPANDING

COMPANDING:-

In telecommunication and signal processing companding (occasionally called compansion) is a method of mitigating the detrimental effects of a channel with limited dynamic range. The name is a portmanteau of compressing and expanding. The use of companding allows signals with a large dynamic range to be transmitted over facilities that have a smaller dynamic range capability. Companding is employed in telephony and other audio applications such as professional wireless microphones and analog recording.



Original signal



After compressing, before expanding

TIME DIVISION MULTIPLEXING:-

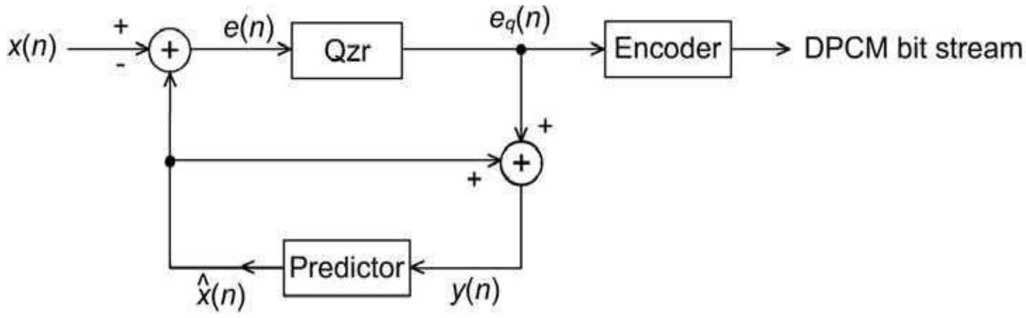
Time-division multiplexing (TDM) is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. This form of signal multiplexing was developed in telecommunications for telegraphy systems in the late 1800s, but found its most common application in digital telephony in the second half of the 20th century.

Differential pulse-code modulation

Differential pulse-code modulation (DPCM) is a signal encoder that uses the baseline of pulse-code modulation (PCM) but adds some functionality based on the prediction of the samples of the signal. The input can be an analog signal or a digital signal.

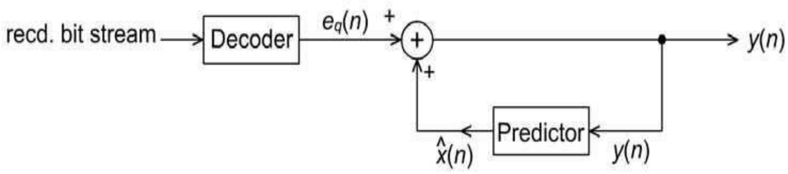
If the input is a continuous-time analog signal, it needs to be sampled first so that a discrete-time signal is the input to the DPCM encoder.

- Option 1: take the values of two consecutive samples; if they are analog samples, quantize them; calculate the difference between the first one and the next; the output is the difference, and it can be further entropy coded.
- Option 2: instead of taking a difference relative to a previous input sample, take the difference relative to the output of a local model of the decoder process; in this option, the difference can be quantized, which allows a good way to incorporate a controlled loss in the encoding Applying one of these two processes, short-term redundancy (positive correlation of nearby values) of the signal is eliminated; compression ratios on the order of 2 to 4 can be achieved if differences are subsequently entropy coded, because the entropy of the difference signal is much smaller than that of the original discrete signal treated as independent samples.



(a)

Fig: (a) DPCM transmitter



(b)

Fig: (b) DPCM receiver

UNIT-03/LECTURE-08

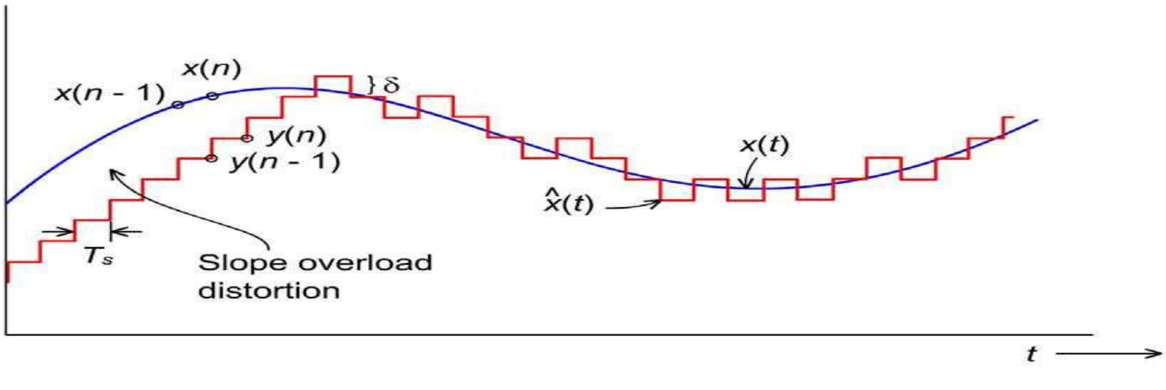
DELTA MODULATION

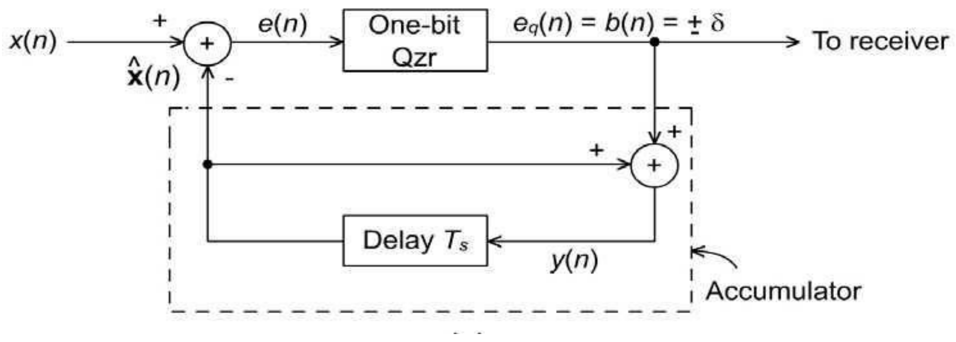
Delta modulation:- **(DEC 2013)(7)**

Delta modulation (DM or Δ-modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance. DM is the simplest form of differential pulse-code modulation (DPCM) where the differences between successive samples are encoded into n-bit data streams. In delta modulation, the transmitted data is reduced to a 1-bit data stream. Its main features are:

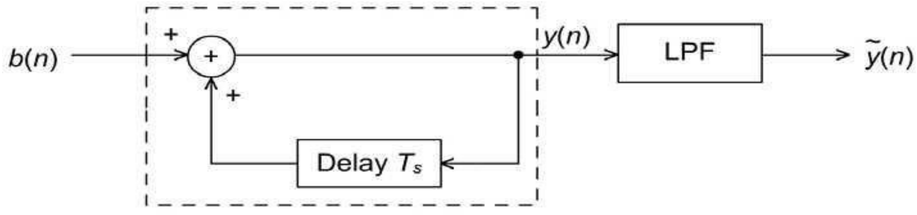
- the analog signal is approximated with a series of segments
- each segment of the approximated signal is compared to the original analog wave to determine the increase or decrease in relative amplitude
- the decision process for establishing the state of successive bits is determined by this comparison
- only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.

Delta modulation, like DPCM is a predictive waveform coding technique and can be considered as a special case of DPCM. It uses the simplest possible quantizer, namely a two level (one bit) quantizer. The price paid for achieving the simplicity of the quantizer is the increased sampling rate (much higher than the Nyquist rate) and the possibility of slope-overload distortion in the waveform reconstruction, as explained in greater detail later on in this section. In DM, the analog signal is highly over-sampled in order to increase the adjacent sample correlation. The implication of this is that there is very little change in two adjacent samples, thereby enabling us to use a simple one bit quantizer, which like in DPCM, acts on the difference (prediction error) signals.
In its original form, the DM coder approximates an input time function by a series of linear segments of constant slope. Such a coder is therefore referred to as a Linear (or non-adaptive) Delta Modulator (LDM). Subsequent developments have resulted in delta modulators where the slope of the approximating function is a variable. Such coders are generally classified under Adaptive Delta Modulation(ADM) schemes. We use DM to indicate either of the linear or adaptive variety.





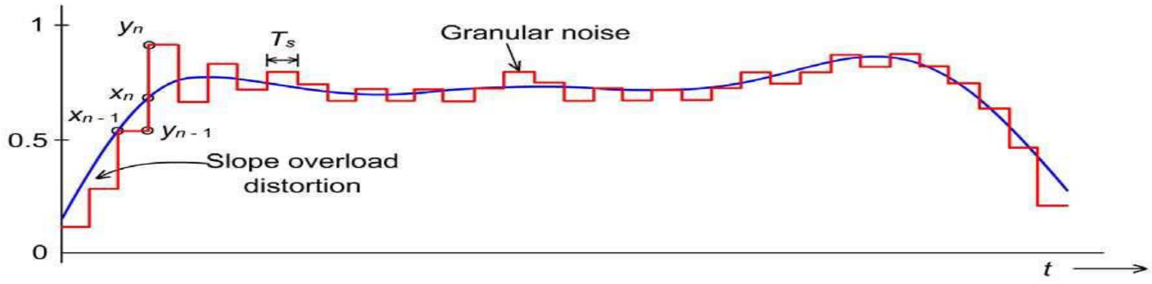
(a)



(b)

Fig.Discrete-time LDM system (a) Transmitter (b) Receiver

fig. indicate two types of quantization errors in DM: slope overload distortion (noise) and granular noise. Slope-overload is said to occur when the step size is too small to follow a steep segment of the input waveform *x(t)*. Granularity, on the other hand, refers to a *x(t)* hunts around a relatively flat segment of the input function, with a step size that is too large relative to the local slope characteristic of the input. It is therefore clear that for a given statistics of the input signal slope relatively small values of accentuate slope-overload while relatively large values of ▯ increase granularity. Slope overload distortion is a basic drawback of the LDM system.



	RGPV QUESTIONS	Year	Marks
Q.1	Describe delta modulation method. What are its limitations? How can they be overcome?	DEC 2013	7
Q.2	Describe delta modulation. What are its limitations? How are they overcome	JUNE 2012	7
Q.3	Explain the advantages and disadvantages of Delta Modulation. How can we overcome this error?	DEC 2010	10

UNIT-03/LECTURE-09
ADAPTIVE DELTA MODULATION

<u>Adaptive delta modulation</u>:- **(DEC 2013)(7)**

Adaptive delta modulation (ADM) or continuously variable slope delta modulation (CVSD) is a modification of DM in which the step size is not fixed. Rather, when several consecutive bits have the same direction value, the encoder and decoder assume that slope overload is occurring, and the step size becomes progressively larger. Otherwise, the step size becomes gradually smaller over time. ADM reduces slope error,at the expense of increasing quantizing error.This error can be reduced by using a low pass filter. ADM provides robust performance in the presence of bit errors meaning error detection and correction are not typically used in an ADM radio design, this allows for a reduction in host processor workload (allowing a low-cost processor to be used).

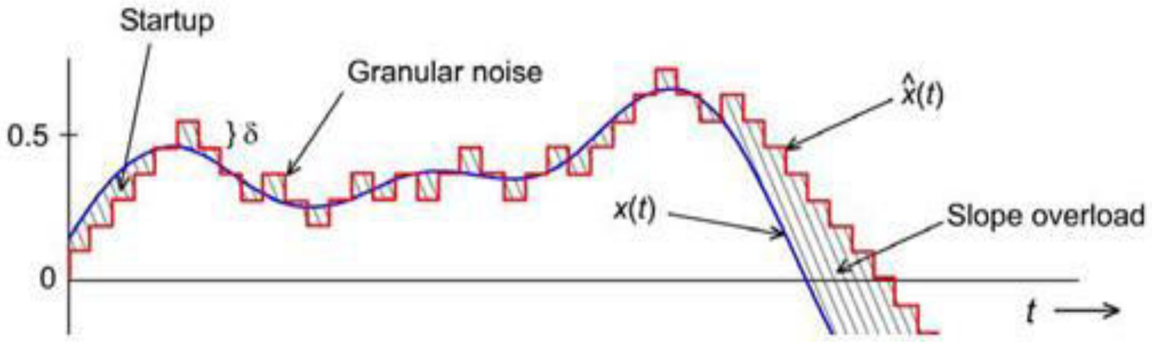
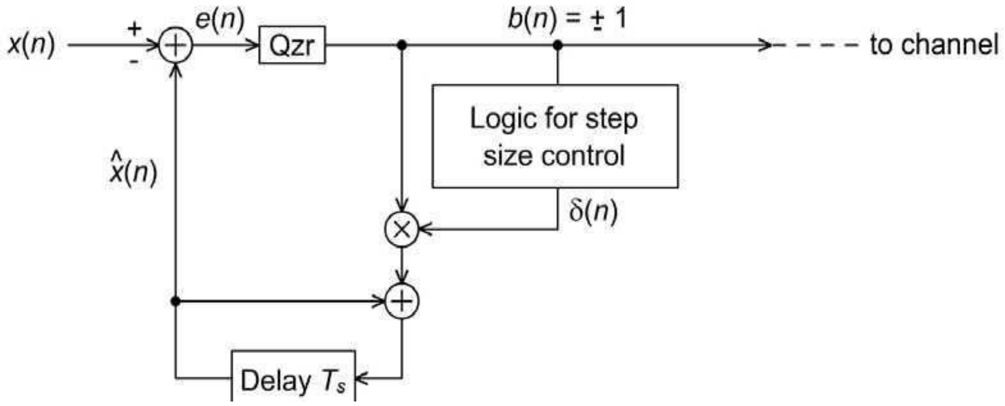


Fig: Waveforms illustrative of ADM operation

Here the step size Δ of the quantizer is not a constant but varies with time. The step size increases during a steep segment of the input and decreases when the modulator is quantizing a slowly

The adaptive step size control which forms the basis of an ADM scheme can be classified in various ways such as: discrete or continuous; instantaneous or syllabic (fairly gradual change); forward or backward.



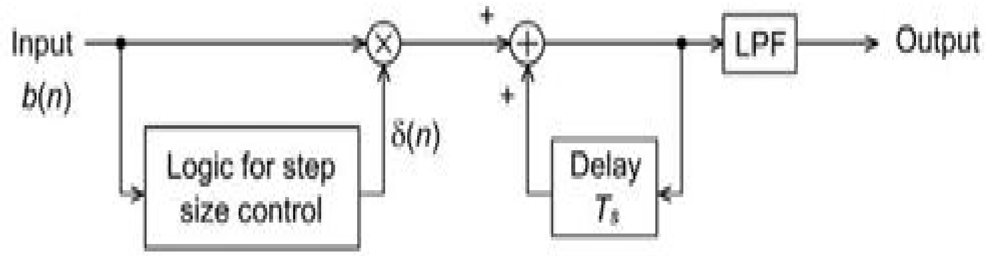


Fig: ADM (a) Transmitter (b) Receiver

for the case of bandpass filtered (200-3200 Hz) speech. For PCM telephony, the sampling frequency used is 8 kHz. As can be seen from the figure, the SNR comparison between ADM and PCM is dependent on the bit rate. An interesting consequence of this is, below 50 kbps, ADM which was originally conceived for its simplicity, out-performs the logarithmic PCM, which is now well established commercially all over the world. A 60 channel ADM (continuous adaptation) requiring a bandwidth of 2.048 MHz (the same as used by the 30 channel PCM system) was in commercial use in France for sometime. French authorities have also used DM equipment for airborne radio communication and air traffic control over Atlantic via satellite. However, DM has not found wide-spread commercial usage simply because PCM was already there first!

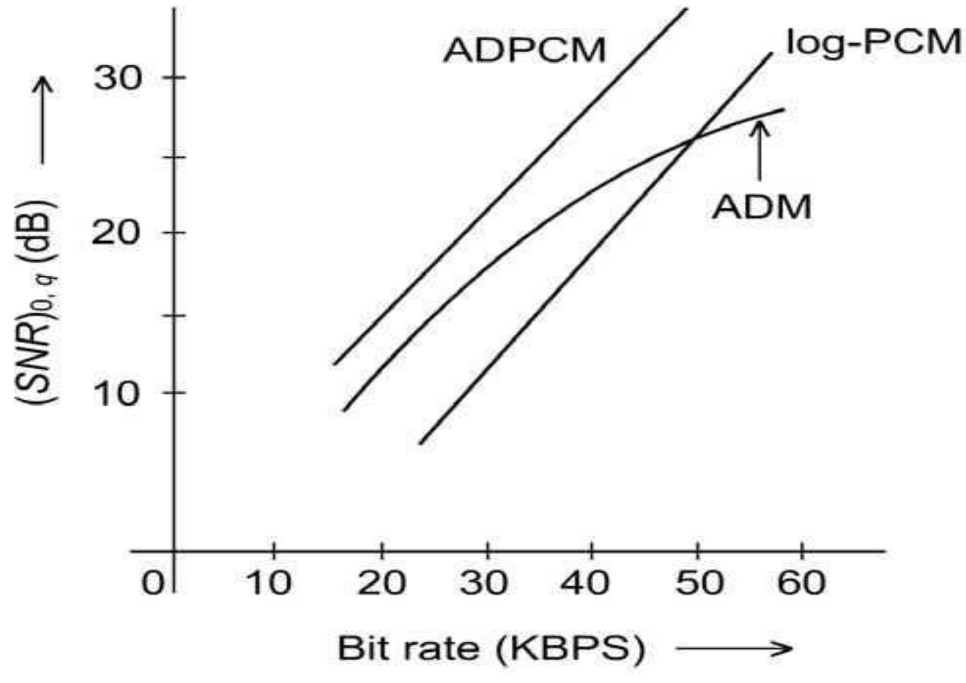


Fig: Performance of PCM and ADM versus bit rate

Instantaneously adapting delta modulators (such as the scheme described above) are more vulnerable to channel noise than the slowly adapting or linear coders. Therefore, although instantaneously adapting delta modulators are very simple and efficient (SNR-wise) in a relatively noise-free environments

Sr. No.	Parameter	PCM	Delta modulation (DM)	Adaptive Delta Modulation (ADM)	Differential Pulse Code Modulation (DPCM)
1.	Number of bits	It can use 4, 8 or 16 bits per sample.	It uses only one bit for one sample.	Only one bit is used to encode one sample.	Bits can be more than one but are less than PCM.
2.	Levels, step size	The number of levels depend on number of bits. Level size is fixed.	Step size is fixed and cannot be varied.	According to the signal variation, step size varies (Adapted).	Fixed number of levels are used.
3.	Quantization error and distortion	Quantization error depends on number of levels used.	Slope overload distortion and granular noise is present.	Quantization error is present but other errors are absent.	Slope overload distortion and quantization noise is present.
4.	Bandwidth of transmission channel	Highest bandwidth is required since number of bits are high.	Lowest bandwidth is required.	Lowest bandwidth is required.	Bandwidth required is lower than PCM.
5	Feedback.	There is no feedback in transmitter or receiver.	Feedback exists in transmitter.	Feedback exists.	Feedback exists.
6	Complexity of notation	System is complex.	Simple.	Simple.	Simple.
7.	Signal to noise ratio	Good.	Poor.	Better than DM.	Fair.
8.	Area of applications	Audio and video Telephony.	Speech and images.	Speech and images.	Speech and video.

COMPARASON BETWEEN PCM,DM,ADM AND DPCM

	RGPV QUESTIONS	Year	Marks
Q.3	Discuss the need of ADM. With the help of block diagram. Explain its working	DEC 2013	7
Q.2	With the help of block diagram explain ADM system.	DEC 2012	7
Q.1	Compare PCM, DM, ADM and DPCM.	DEC 2011	10